IN THE CLAIMS:

This listing of the claims replaces all prior versions and listings of the claims.

Claim 1. (currently amended) A method for generating speech recognition models, the method comprising:

converting speech spoken from a plurality of female speakers into a first set of recorded phonemes training data;

converting speech spoken from a plurality of male speakers into a second set of recorded phonemes training data;

receiving a first speech recognition model based on ~~a~~ the first set of recorded phonemes training data~~, the first set of recorded phonemes training data originating from a plurality of female speakers~~;

receiving a second speech recognition model based on ~~a~~ the second set of recorded phonemes training data~~, the second set of recorded phonemes training data originating from a plurality of male speakers~~;

determining a difference in model information between the first speech recognition model and the second speech recognition model; and

creating a gender-independent speech recognition model based on the first set of recorded phonemes training data and the second set of recorded phonemes training data if the difference in model information is insignificant.

Claim 2. (original) The method of claim 1, wherein whether the model information is insignificant is based on a threshold model quantity.

Claim 3. (previously presented) The method of claim 1, wherein determining the difference in model information includes calculating a Kullback Leibler distance between the first speech recognition model and second speech recognition model.

Claim 4. (original) The method of claim 3, wherein whether the model information is insignificant is based on a threshold Kullback Leibler distance quantity.

Claim 5. (previously presented) The method of claim 1, wherein the first speech recognition model, second speech recognition model, and gender-independent speech recognition model are Gaussian mixture models.

Claim 6. (currently amended) A system for generating speech recognition models, the ~~method~~ system comprising:

a computer processor;

a first speech recognition model based on a first set of training data, the first set of training data originating from a first set of common entities;

a second speech recognition model based on a second set of training data, the second set of training data originating from a second set of common entities; and

a processing module configured to create an independent speech recognition model based on the first set of training data and the second set of training data if the difference in model information between first speech recognition model and the second speech recognition model is insignificant.

Claim 7. (original) The system of claim 6, wherein whether the model information is insignificant is based on a threshold model quantity.

Claim 8. (previously presented) The system of claim 6, wherein the processing model is further configured to calculate a Kullback Leibler distance between the first speech recognition model and second speech recognition model.

Claim 9. (original) The system of claim 8, wherein whether the model information is insignificant is based on a threshold Kullback Leibler distance quantity.

Claim 10. (currently amended) The ~~method~~ system of claim 6, wherein the first speech recognition model, second speech recognition model, and independent speech recognition model are Gaussian mixture models.

Claim 11. (previously presented) A computer program product embodied in computer memory comprising:

computer readable program codes coupled to the computer memory for generating speech recognition models, the computer readable program codes configured to cause the program to:

receive a first speech recognition model based on a first set of training data, the first set of training data originating from a first set of common entities;

receive a second speech recognition model based on a second set of training data, the second set of training data originating from a second set of common entities;

determine a difference in model information between the first speech recognition model and the second speech recognition model; and

create an independent speech recognition model based on the first set of training data and the second set of training data if the difference in model information is insignificant.

Claim 12. (original) The computer program product of claim 11, wherein whether the model information is insignificant is based on a threshold model quantity.

Claim 13. (original) The computer program product of claim 11, wherein determining the difference in model information includes calculating a Kullback Leibler distance between the first model and second model.

Claim 14. (original) The computer program product of claim 13, wherein whether the model information is insignificant is based on a threshold Kullback Leibler distance quantity.

Claim 15. (previously presented) The computer program product of claim 11, wherein the first speech recognition model, second speech recognition model, and independent speech recognition model models are Gaussian mixture models.

Claim 16. (currently amended) A system for generating speech recognition models, the method comprising:

<u>a computer processor;</u>

a first speech recognition model based on a first set of training data, the first set of training data originating from a first set of common entities;

a second speech recognition model based on a second set of training data, the second set of training data originating from a second set of common entities; and

means for creating an independent speech recognition model based on the first set of training data and the second set of training data if the difference in model information between first speech recognition model and the second speech recognition model is insignificant.

Claim 17. (currently amended) A method for recognizing speech from an audio stream originating from one of a plurality of data classes, the method comprising:

<u>converting the speech into the audio stream;</u>

receiving a current feature vector of the audio stream;

computing a current vector probability that the current feature vector belongs to one of the plurality of data classes;

computing an accumulated confidence level that the audio stream belongs to one of the plurality of data classes based on the current vector probability and on previous vector probabilities;

weighing class models based on the accumulated confidence; and

recognizing the current feature vector based on the weighted class models; and

wherein the plurality of data classes include a female speech recognition model based on recorded phonemes originating from plurality of female speakers, a male speech recognition model based on recorded phonemes originating from plurality of male speakers, and a gender-independent speech recognition model based on recorded phonemes originating from plurality of both female and male speakers having insignificant differences in information.

Claim 18. (original) The method of claim 17, wherein computing the current vector probability includes estimating an a posteriori class probability for the current feature vector.

Claim 19. (original) The method of claim 17, wherein computing the accumulated confidence level further comprising weighing the current vector probability more than the previous vector probabilities.

Claim 20. (original) The method of claim 17, further comprising determining if another feature vector is available for analysis.

Claim 21. (currently amended) A system for recognizing speech data from an audio stream originating from one of a plurality of data classes, the system comprising:

a computer processor;

a receiving module configured to receive a current feature vector of the audio stream;

a first computing module configured to compute a current vector probability that the current feature vector belongs to one of the plurality of data classes;

a second computing module configured to compute an accumulated confidence level that the audio stream belongs to one of the plurality of data classes based on the current vector probability and on previous vector probabilities;

a weighing module configured to weigh class models based on the accumulated confidence; and

a recognizing module configured to recognize the current feature vector based on the weighted class models; and

wherein the plurality of data classes include a first speech recognition model based on recorded phonemes originating from a first set of speakers, a second speech recognition model based on recorded phonemes from a second set of speakers, and a third speech recognition model based on recorded phonemes originating from both the first and second set of speakers having insignificant differences in information.

Claim 22. (original) The system of claim 21, wherein the first computing module is further configured to estimate an a posteriori class probability for the current feature vector.

Claim 23. (original) The system of claim 21, wherein the second computing module is further configured to weigh the current vector probability more than the previous vector probabilities.

Claim 24. (previously presented) A computer program product embodied in computer memory comprising:

computer readable program codes coupled to the computer memory for recognizing speech data from an audio stream originating from one of a plurality of data classes, the computer readable program codes configured to cause the program to:

receive a current feature vector of the audio stream;

compute a current vector probability that the current feature vector belongs to one of the plurality of data classes;

compute an accumulated confidence level that the audio stream belongs to one of the plurality of data classes based on the current vector probability and on previous vector probabilities;

weigh class models based on the accumulated confidence; and

recognize the current feature vector based on the weighted class models; and

wherein the plurality of data classes include a first speech recognition model based on recorded phonemes originating from a first set of speakers, a second speech recognition model based on recorded phonemes from a second set of speakers, and a third speech recognition model based on recorded phonemes

originating from both the first and second set of speakers having insignificant differences in information.

Claim 25. (original) The computer program product of claim 24, wherein the program code configured to compute the current vector probability includes program code configured to determine an a posteriori class probability for the current feature vector.

Claim 26. (original) The computer program product of claim 24, wherein the program code configured to compute the accumulated confidence level includes program code configured to weigh the current vector probability more than the previous vector probabilities.

Claim 27. (original) The computer program product of claim 24, further comprising program code configured to determine if another feature vector is available for analysis.